REMARKS

In response to the Office Action dated August 13, 2007, Applicants hereby request reconsideration of the pending claims in light of the amendments above and the following remarks.

STATUS OF CLAIMS

Claims 1-22 as originally filed were pending.

Claims 5, 6, and 10-22 are canceled.

Claims 23-29 are newly added.

Accordingly, claims 1-4, 7-9, and 23-29 are before the Examiner for consideration.

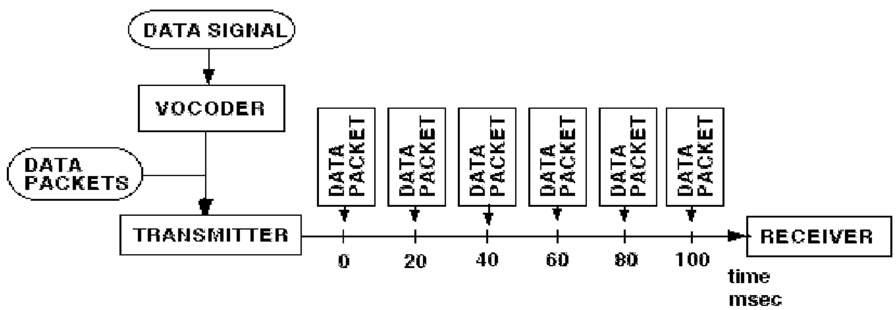
CLAIM REJECTIONS

In section 2 of the Office Action, claims 1-8, 10-18, and 20-22 were rejected under 35 U.S.C. § 103(a) as being unpatentable over U.S. Pat. No. 7,002,993 to Mohaban et al. ("Mohaban") in view of U.S. Application Publication No. 2003/0091017 to Davenport ("Davenport"). Claims 5, 6, 10-18, and 20-22 are canceled in favor of new claims 23-29. Applicants respectfully traverse the subject rejection in regards to claims 1-4, 7, and 8, since Mohaban and Davenport fail to show or suggest in combination all the elements of these claims as amended.

A partial summary of the packet aggregation system of the present application can be found in Applicants' Response dated July 12, 2007.

Turning first to independent claim 1, claim 1 has been amended to relate to an embodiment of the packet aggregation system summarized generally in sections 0010-0013 of the application. Here, the packet aggregation system is used in the context of transmitting time-delay intolerant information (also known as real-time information), which is information having one or more time strict time delay constraints. As explained in section 0004-0008 and 0022 of the application, for real-time packet data applications, an information stream (e.g., a digitized voice signal)

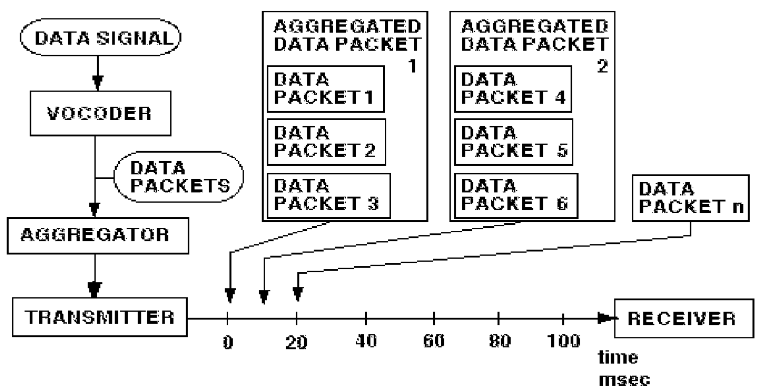
is converted into a stream of time-delay intolerant data packets, which are scheduled for transmission at a designated fixed rate depending on the time delay constraint(s) of the information in question. In a conventional system, the data packets are then periodically transmitted from a transmitter to a receiver at the fixed rate. This process is shown in the sketch below, for the case of an EVRC (Enhanced Variable Rate Codec) vocoder.



Here, the EVRC vocoder converts a voice/sound data signal into a stream of data packets, which are scheduled for transmission at a designated fixed rate of about one packet every 20 milliseconds. As indicated, a transmitter then transmits the data packets at the fixed rate, for reception at a receiver unit. Thus, a first packet is transmitted at time = 0 seconds, a second at time = +20 milliseconds, a third at time = +40 milliseconds, and so on. The fixed rate of 1 packet/20 msec is based on the time constraints of the voice data signal/information. For example, in a typical communication system it is necessary to transmit a voice data packet (containing 16-192 bits of information) every 20 milliseconds in order to adequately reproduce the original voice data signal at the receiver. (Transmitting packets at a lower rate may result in poor overall sound quality, breaks in sound, and the like.)

The packet aggregation system of the present application is directed to data aggregation in this context, for aggregating time-delay intolerant information. Thus, as summarized in sections 0010-0013 of the application, time-delay intolerant data packets, which are scheduled for transmission at a designated rate, are generated as explained above, e.g., using a vocoder or other signal encoder. Then, however, instead of transmitting the data packets

at the designated rate, e.g., 1 packet every 20 msec, two or more of the time-delay intolerant data packets may be aggregated into an aggregated data packet, if channel conditions warrant, and based on other user service requirements and a negotiation between the transmitter and receiver. The aggregated data packet is then transmitted over the communication channel at a rate different than the designated rate of the original time-delay intolerant data packets.



An example of the system in operation is shown in the sketch above. Here, a stream of time-delay intolerant data packets (i.e., "real-time" data packets having strict time constraints), such as generated by a vocoder unit, is provided to an aggregator. Based on channel conditions of the communication channel over which the data is to be transmitted, and possibly on user service requirements such as overall system load, the data rate capability (e.g., bandwidth) of the system/channel, and a QoS (quality of service) level assigned to the receiver, the transmitter negotiates with the receiver to determine an aggregation factor "N," which is the number of data packets to be aggregated into an aggregated data packet. If the system is in a state of heavy load, or if the channel conditions are poor, then it may be the case that N = 1, in which case no data packets are aggregated, and the original data packets are simply transmitted at the designated rate, as normal. If however the system is in a state of light load, or if channel conditions are particularly good, then N > 1 data packets are aggregated into an aggregated data packet, which can be transmitted at a rate other than the designated rate. As shown in the sketch above, for example, three data packets are aggregated

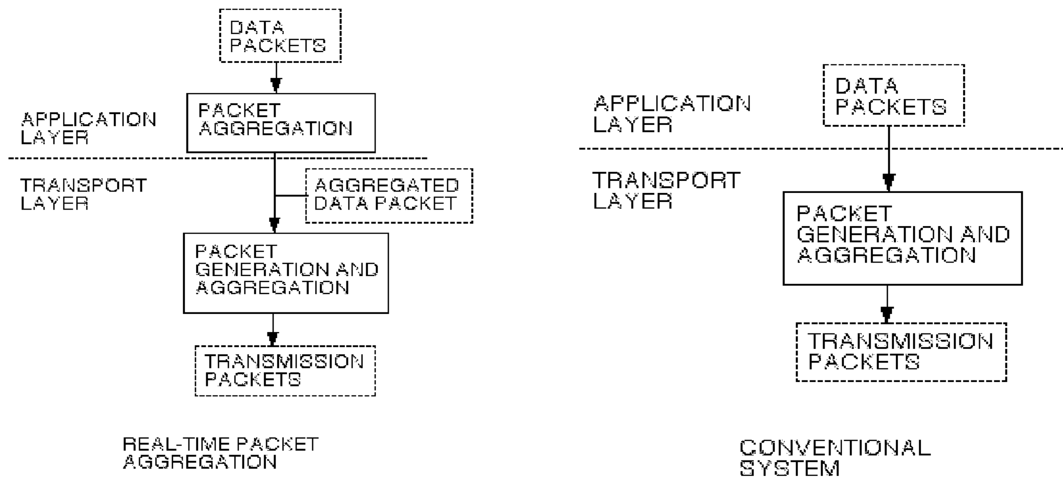
into a first aggregated data packet for transmission at time = 0, and three more data packets are aggregated into a second aggregated data packet for transmission at time = +10 msec. As should be appreciated, one goal of this method is to get more data packets to the receiver faster, when conditions warrant, than simply waiting for the data packets to be transmitted at the designated rate. Not only does the aggregation process improve communication efficiency generally, but if fewer data packets are transmitted at the fixed rate, then it is possible to add more users to the transmission queue, i.e., additional time slots are available for scheduling more users.

As an example, one situation where this method/system is particularly advantageous is in a wireless network during an emergency or other heavy load period. At the onset of the emergency, say a major snowstorm, a number of mobile phone users may attempt to access the network at the same time, to contact family members, work, and emergency and other service providers. Since each user is generating real-time voice signals, which are scheduled for transmission at a fixed rate, in a conventional system the number of available time/data slots will quickly fill up. In other words, at each time point, the number of data packets scheduled for transmission may exceed the available transmission bandwidth. If this happens, some users may be denied service. In the packet aggregation system of the present application, however, channel conditions are monitored (in addition to other factors mentioned above) for determining if it is possible to aggregate time-delay intolerant data packets for a particular user. If so, "N" packets are aggregated into an aggregated data packet, as described above, which is transmitted at a different rate than the designated or fixed rate. In the sketch above for example, three data packets are aggregated into the first aggregated data packet, for transmission at time = 0. Thus, instead of the three data packets occupying three time slots, they occupy a single time slot. The other time slots can then be used to accommodate other users.[1]

---

[1] The bandwidth used by one aggregated packet is greater than a single data packet, but less than if the multiple packets were transmitted separately. Plus, there are advantages in being able to accommodate more users due to the increase in available time slots for transmitting time-delay intolerant information.

It should be noted that in the context of aggregating time-delay intolerant information as described above, Applicants' packet aggregation system inherently works in the application layer of the communication system. (According to the standard TCP/IP model or Internet reference model, the application layer is the top network layer, which is used by most programs for network communication. Data is passed from the program in an application-specific format, then encapsulated into a transport layer protocol. Common application layer programs include file transfer and management protocols, directory services, and communication services/protocols such as HTTP. Below the application layer are the transport layer, the network layer, the data link layer, and the physical layer.) Thus, aggregation of time-delay intolerant data packets is carried out as an application process, e.g., vocoder voice signal conversion and transmission. Because packets are aggregated at the application layer, it inherently follows that the aggregated data packets will be further processed according to the communication system's protocols at the lower layers of system functionally. In the packet aggregation system of the present application, which operates in the context of a packet data communication system/network, therefore, an aggregated data packet may be re-divided into packets, at the transport layer, for transmission over the communication channel of interest. The transmission packets do not correspond to the original time-delay intolerant data packets. Instead, they are formatted according to the network's transport layer protocols, just as if the aggregated data packet was "one big" time-delay intolerant data packet. As should be appreciated, therefore, for aggregating time-delay intolerant data packets, the real-time packet aggregation system of the present application works above and separately from transmission packet generation and aggregation at the transport layer, i.e., the transport layer may perform its own packet aggregation, but of transmission packets, not of time-delay intolerant data packets. This difference is illustrated in the sketch below.



Claim 1 is directed to a transmitting communication equipment used in a system as described above, for aggregating time-delay intolerant information. Claim 1 includes the following elements (paraphrased):

- An aggregator for aggregating information and for transmitting the aggregated information as an aggregated packet to a receiving communication equipment.
- The information comprises a plurality of time-delay intolerant data packets scheduled for transmission at a designated rate. The designated rate is based on at least one time delay constraint of the data packets.
- An aggregation factor, which is the number of data packets included in the aggregated packet, is based at least in part on (i) one or more user service requirements and (ii) a negotiation between the transmitting communication equipment and the receiving communication equipment.
- The aggregated packet is transmitted by the transmitting equipment to the receiving equipment at a rate different than the designated rate of the data packets.

None of the references of record, alone or in combination, show all of these elements. More specifically, neither Mohaban nor Davenport show a system where time-delay intolerant data packets are aggregated into an aggregated data packet, where the size of the aggregated packet is negotiated between

the transmitter and receiver and based on user service requirements, and where the aggregated packet is transmitted at a rate different than the scheduled transmission of the time-delay intolerant data packets.

As explained in the previous Response, Mohaban relates to aggregating media data packets from different phone calls or other transmissions, for improving bandwidth efficiency. Aggregated data packets are generated using compressed RTP (real-time transport protocol) segments and a set of ID codes (e.g., a trunk ID and session context ID) to identify the individual data packets in the aggregated packet. See Mohaban at col. 3, lines 34-49. Considering that RTP is a transport layer mechanism (in the TCP/IP model), and in light of the explanation above, the system in Mohaban can be thought of as a transport layer aggregation method, i.e., Mohaban involves modifying an existing transport layer mechanism to improve bandwidth efficiency.[2] Generation of the aggregated packets is a standalone operation, and does not involve considerations of user service requirements or a negotiation between the transmitting and receiving equipment. Moreover, there is no mention in Mohaban of aggregating time-delay intolerant data packets into an aggregated data packet, which is then transmitted at a different rate than the scheduled transmission rate of the time-delay intolerant data packets.

The Davenport reference fails to rehabilitate the deficiencies of Mohaban. In particular, in the Office Action at page 4, the Examiner agreed that Mohaban does not teach basing the size of an aggregated data packet on a negotiation between the transmitting and receiving communication equipment, but went on to opine that Davenport teaches basing the size of a packet on such a negotiation. Applicants agree that Davenport discloses adjusting packet size, and that "the quality of the communication link … may be considered to determine the optimal packet size." See Davenport at

---

[2] More specifically, Mohaban is concerned with the application layer, but achieves aggregation by modifying the underlying IP/transport layer through introduction of a custom header. Compression may be included as well.

section 0019.[3] However, there is no explicit teaching in Davenport of a negotiation between the transmitting and receiving equipment. Moreover, to the extent Davenport suggests such an operation, it is important to note that Davenport does not relate to packet aggregation in the first place. Instead, in Davenport a "chunk" of data is broken into packets, the size of which may be adjusted based on the quality of the communication link; there is no mention or suggestion in Davenport of aggregating data packets. As such, even if "Davenport teaches a size of the packet is based at least in part on a negotiation between the transmitting communication equipment and the receiving communication equipment," as stated by the Examiner, Davenport does not teach basing the size of an *aggregated* packet on such a negotiation, where the size is a direct function of the number of data packets included in the aggregated packet. Put another way, to the extent Davenport suggests a transmitter/receiver negotiation, this would be used to determine how to divide a larger data block into smaller packets. Applicants' packet aggregation system, on the other hand, relates to basing the number of smaller packets in a larger, aggregated packet on a transmitter/receiver negotiation. The two processes are not interchangeable, and a disclosure of using a transmitter/receiver negotiation as a factor in one system does not translate as a teaching applicable to the other system. The same is true in regards to communication channel conditions – the teaching in Davenport of basing packet size on communication link quality is different from basing the size of an aggregated packet (e.g., the number of data packets in the aggregated packet) on communication link quality, as in Applicants' system.

In the Office Action, the Examiner states that Davenport explicitly teaches determining the size of an aggregated data packet through a transmitter/receiver negotiation. See, e.g., Office Action at page 8. Applicants respectfully disagree. Instead, Davenport teaches that optimal packet size may be determined using communication link quality, when dividing bulk transfer data into packets for transmission. See, e.g., Davenport

[3] Based on Applicants' review and understanding of Davenport, it appears that the packet generation process disclosed in Davenport is of the type typically used in communication networks, e.g., packet size is based on communication channel conditions, which is a known, standard process.

at sections 0018-0019 ("Once communication has been initiated 48 between the mobile asset and a fixed location via a communication link, the size of the data packets used to download the data may be selected, as is known in the art of digital communications.") Dividing data into packets for transmission is not the same as aggregating data packets into an aggregated data packet.

To the extent the use of the term "packet size" in claim 1 could be construed in a manner for falling within the ambit of Davenport, claim 1 has been amended to recite that that aggregation factor, which is the number of data packets included in the aggregated packet, is based on a negotiation between transmitter and receiver. Since the system in Davenport does not utilize or otherwise relate to packet aggregation (indeed, quite the opposite), it cannot reasonably be said that Davenport teaches basing an aggregation factor on a transmitter/receiver negotiation.

Further evidence that Davenport is far removed from the packet aggregation system of the present application can be found by taking note of the type of data transferred in Davenport. In particular, Davenport relates to bulk data transfer, for obtaining systems or operational data from a train or other moving vehicle. See Davenport at sections 0003, 0012, 0017, etc., where it is noted that (i) the system in Davenport is unrelated to voice (i.e., real-time) data transfer, and is instead related to bulk data transfer, and (ii) data transfer is typically scheduled in advance. Thus, even if Davenport taught packet aggregation, and the idea of basing the number of bulk transmission data packets in an aggregated packet on channel conditions and/or a transmitter/receiver negotiation (which is not the case), then there would still be no teaching or suggestion of aggregating time-delay intolerant data packets into an aggregated data packet, or of basing the number of time-delay intolerant data packets in the aggregated packet on channel conditions and/or a transmitter/receiver negotiation, in a real-time application.

On a side note, in the instances where Davenport mentions "real-time" processes (see, e.g., Davenport at section 0023), this relates to real-time

measurements of channel conditions, not to the processing of real-time packet or other data.

In light of the above, it is respectfully submitted that Mohaban and Davenport in combination fail to disclose all the elements of claim 1 as amended. Accordingly, claim 1 is believed allowable. Claims 2-4 and 7-9, which depend from claim 1, are believed allowable as depending from an allowable base claim, and for reciting additional elements neither shown nor suggested in Davenport and Mohaban. For example:

- Claim 2 recites that the aggregation factor (i.e., the number of time-delay intolerant data packets in the aggregated data packet) is based on "channel conditions of a communication channel used for transmitting the aggregated packet between the transmitting communication equipment and the receiving communication equipment." Mohaban relates to aggregating unrelated packets by modifying the RTP codec, and does not teach aggregating packets based on channel conditions. As discussed above, Davenport teaches basing packet size on channel conditions (in the context of dividing bulk data into packets), but does not disclose generating aggregated packets, or basing the number of packets in an aggregated packet on channel conditions.
- Claim 3 is further allowable for reasons similar to those set forth immediately above in regards to claim 2.
- In regards to claim 4, neither Davenport nor Mohaban relate to transmitting or otherwise processing time-delay intolerant data packets, scheduled for transmission at a designated rate, at a rate other than the designated rate.
- Claim 7 recites that the aggregation factor is further based on user service requirements that include "(i) a data rate capability of a communication system within which the equipment is being used, (ii) a current loading level of the system and/or channel, and (iii) a designated quality of service level of the receiving communication equipment in the system." (The latter factor is described in section 0026 of the application.) Davenport discloses using channel conditions

in an unrelated context, but neither Davenport nor Mohaban disclose basing an aggregation factor on such user service requirements.

In section 3 of the Office Action, claims 9 and 19 were rejected under 35 U.S.C. § 103(a) as being unpatentable over Mohaban in view of Davenport and further in view of U.S. Pat. No. 6,839,356 to Barany et al. Claim 19 is canceled. Claim 9 is believed allowable as depending from an allowable base claim, since Mohaban and Davenport in combination fail to show all the elements of the claims from which claim 9 depends.

To the extent any grounds for rejection were not explicitly addressed herein, they are believed moot in light of the claim amendments above.

NEW CLAIMS

Newly entered claims 23-29 are believed allowable for reasons similar to those set forth above in regards to claims 1-4, 7, and 8. New claims 25 and 29 recite elements inherent to the disclosure of the application, as discussed above. Applicants aver that no new matter is entered.

CONCLUSION

In view of the foregoing, it is respectfully submitted that pending claims 1-4, 7-9, and 23-29 are in condition for allowance, and action to that effect is earnestly solicited.

Pursuant to 37 C.F.R. § 1.136, Applicants hereby petition for a one-month of extension of time for response, thereby extending the period for response to and through December 13, 2007. Authorization is hereby given to charge any fees owed to our Deposit Account No. 13-0235.

Respectfully submitted,

By /John A. Kramer/
John A. Kramer
Registration No: 46,302
Attorney for Applicants

McCORMICK, PAULDING & HUBER LLP
CityPlace II
185 Asylum Street
Hartford, CT 06103-3402
(860) 549-5290
Customer No.: 51094